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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/731,084	12/06/2000	Jon A. Arrowood	8999	9387

26890 7590 02/10/2005

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EXAMINER

CHAWAN, VIJAY B

ART UNIT

PAPER NUMBER

2654

DATE MAILED: 02/10/2005

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**MAILED**  
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**BEFORE THE BOARD OF PATENT APPEALS  
AND INTERFERENCES**

Application Number: 09/731,084  
Filing Date: December 06, 2000  
Appellant(s): ARROWOOD ET AL.

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James M. Stover  
For Appellant

**EXAMINER'S ANSWER**

This is in response to the appeal brief filed July 14, 2004.

**(1) *Real Party in Interest***

A statement identifying the real party in interest is contained in the brief.

**(2) *Related Appeals and Interferences***

A statement identifying the related appeals and interferences which will directly affect or be directly affected by or have a bearing on the decision in the pending appeal is contained in the brief.

**(3) *Status of Claims***

The statement of the status of the claims contained in the brief is correct.

**(4) *Status of Amendments After Final***

The appellant's statement of the status of amendments after final rejection contained in the brief is correct.

The amendment after final rejection filed on 5/12/04 has been entered upon appeal.

**(5) *Summary of Invention***

The summary of invention contained in the brief is correct.

**(6) *Issues***

The appellant's statement of the issues in the brief is correct.

**(7) Grouping of Claims**

Appellant's brief includes a statement that claims 3 through 6 stand or fall together and provides reasons as set forth in 37 CFR 1.192(c)(7) and (c)(8).

**(8) Claims Appealed**

The copy of the appealed claims contained in the Appendix to the brief is correct.

**(9) Prior Art of Record**

5,574,824	Slyh et al.	11-1996
6,009,396	Nagata	12-1999
6,061,646	Martino et al.	5-2000

**(10) Grounds of Rejection**

The following ground(s) of rejection are applicable to the appealed claims:

***Claim Rejections - 35 USC § 103***

Claims 3-4 are rejected under 35 U.S.C. 103(a) as being unpatentable over Martino et al., (6,061,646) in view of Slyh et al., (5,574,824) and in view of Nagata (6,009,396).

As per claim 3, Martino et al., teach an apparatus comprising a self service kiosk, which dispenses articles, currency, or communication services (Col.1, lines 41-60, Fig.2).

Martino et al., do not specifically teach, that within the kiosk, a steerable beam microphone array having multiple lobes, and means for sampling lobes, and distinguishing the difference between speech content and noise content from sound signals received from each lobe. Slyh et al., teach a steerable beam microphone array (Col.2, lines 44-45). Slyh et al., also teach distinguishing the difference between speech content and noise content from sound signals (Col.5, lines 40-41, Col.6, lines 32-37, SNR- Signal-to-Noise Ratio). Signal-to-Noise Ratio implies that the signal power and the noise power have been separately computed. In this case, the signal power is construed to be speech signal power. Therefore it would have been obvious to one with ordinary skill in the art at the time of invention to incorporate the beam microphone array as taught by Slyh et al., in the Kiosk for multiple spoken languages of Martino et al., because, this would provide the user with an improved system using a microphone array to enhance speech that has been corrupted by several directional interference signals and/or additive background noise.

Martino et al., in view of Slyh et al., do not specifically teach identifying lobes having a relatively low noise content, or relatively high speech content, and actuating a lobe having both a relatively high speech content and relatively low

noise content. Nagata et al., in the same field of endeavor, teach a sound source position search unit that estimates a power arriving from each position (Col.6, lines 1-2, and 42-47, Col.9, lines 29-35, Figs., 3 and 6). The sound source position search unit is the equivalent of ii) means for sampling lobes, since as described in the specification, a lobe is a plot of magnitude versus angular position. Nagata et al., teach that all peaks above a threshold are detected as sound sources (Col.10, lines 4-5). This is equivalent to identifying lobes having a relatively low noise content, i.e., detecting high speech content. Nagata et al., also teach a speech parameter extraction unit that extracts the power for each bandwidth and uses it as a speech parameter, which in turn is sent to the speech recognition unit (Col.10, lines 24-27, Fig.3). In the speech recognition unit, the speech power is calculated from the speech parameter (Col.10, lines 32-33), i.e., identifying a signal with high speech content, and low noise content.

Therefore, it would have been obvious to one with ordinary skill in the art at the time of invention to incorporate detecting speech content in a signal as taught by Nagata et al., in the apparatus of Martino et al., in view of Slyh et al., because, an artisan would readily recognize that it would effectively place the beam of the microphone for higher degree of speech recognition, and would effectively suppress background noise and localize sound sources within a kiosk with a steerable microphone array having multiple lobes.

As per claim 4, Martino et al., in view of Slyh et al., and in view of Nagata et al., teach the apparatus as per claim 3, further comprising speech recognition means for recognizing speech contained in the lobe actuated. Nagata et al., teach the band-pass power of the sound source obtained sent from the speech parameter extraction unit to the speech recognition unit and used in the speech recognition processing (Col.10, lines 24-31, Fig.7). Therefore, it would have been obvious to one with ordinary skill in the art at the time of invention to include the means for recognizing speech contained in the lobe of a steerable microphone array, because one with ordinary skill in the art would recognize that this would process only that part of signal that has high speech content and low noise content for more precise speech recognition capability.

Claims 5 and 6 are rejected under 35 U.S.C. 103(a) as being unpatentable over Martino et al., (6,061,646) in view of Nagata et al., (6,009,396).

As per claim 5, Martino et al., disclose a method comprising the following step: maintaining a self-service kiosk which dispenses articles, currency, or communication services (Col.1, lines 41-60, Fig.2). However, Martino et al., fail to teach: maintaining a beam-steerable microphone array at the self-service kiosk, measuring noise content and speech content of several lobes of the array, and, selecting the lobe which carries, larger speech signals than other lobes, and smaller noise signals than other lobes. Nagata et al., teach a speech recognition system

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using a microphone array (Col.1, lines 37-44, Fig.1). Nagata et al., also teach that all the peaks on the sound source distribution above a predetermined threshold are detected as sound sources (Col.10<sup>1</sup>, lines 4-5). One with ordinary skill in the art would be able to measure noise content of the several lobes of the array, since Nagata et al., already distinguishes noise from sound in the signal coming from the microphone array, i.e., the threshold can be considered as noise floor, when the signal is above this threshold, then it is considered to be speech and the parameters extracted, and below is noise. Also, Nagata et al., teach a speech parameter extraction unit that extracts the power for each bandwidth and uses it as a speech parameter. This speech parameter is then sent to the speech recognition unit (Col.10, lines 24-27, Fig.3), in which speech signal power is calculated from the extracted speech parameter (Col.10, lines 32-33). Therefore, it would have been obvious to one with ordinary skill in the art at the time of invention to modify the method of Martino et al., to further comprise maintaining a beam-steerable microphone array at the self-service kiosk, with the method taught by Nagata et al., of measuring noise content and speech content of several lobes of the array, and selecting a lobe which carries larger speech signals than other lobes and smaller noise signals than other lobes, because one of ordinary skill in the art at the time of invention would readily recognize that this would provide more accurate speech recognition by suppressing background noise and localizing sound sources effectively. Also, it would have been obvious to one with ordinary skill in



the art to realize that the combination of Martino et al., in view of Nagata et al., that the resultant filter configuration with a plurality of delay line taps is used so that the power in each direction or position is obtained for each frequency bandwidth necessary for the speech recognition, rather than obtaining the power in each direction or position for each frequency that was used conventionally, because the obtained power can be directly used for the speech recognition while the required amount of calculations used is greatly reduced (Nagata et al., Col.14, lines 60-67).

As per claim 6, Martino et al., in view of Nagata et al., teach the method as per claim 5. Nagata et al., teach receiving signals from the lobe selected, and performing speech recognition on the data. Nagata et al., teach a speech recognition unit whereby speech power is calculated from the speech parameters extracted by the speech parameter extraction unit, and a speech section detected by a speech section detection unit according to the speech power. Then a pattern matching unit carries out pattern matching with a recognition dictionary so that speech recognition is realized (Col.10, lines 32-39). Therefore, it would have been obvious to one with ordinary skill in the art at the time of invention, to modify the method of Martino et al., to further comprise the step of receiving signals from the lobe selected, and performing speech recognition on the data, because one of ordinary skill in the art would readily recognize that this would allow speech

recognition on a selected part of where the signal is most likely carried as opposed to noise.

**(11) Response to Argument**

In response to applicant's argument that the examiner's conclusion of obviousness is based upon improper hindsight reasoning, it must be recognized that any judgment on obviousness is in a sense necessarily a reconstruction based upon hindsight reasoning. But so long as it takes into account only knowledge which was within the level of ordinary skill at the time the claimed invention was made, and does not include knowledge gleaned only from the applicant's disclosure, such a reconstruction is proper. See *In re McLaughlin*, 443 F.2d 1392, 170 USPQ 209 (CCPA 1971).

In response to applicant's argument that there is no suggestion to combine the references, the examiner recognizes that obviousness can only be established by combining or modifying the teachings of the prior art to produce the claimed invention where there is some teaching, suggestion, or motivation to do so found either in the references themselves or in the knowledge generally available to one of ordinary skill in the art. See *In re Fine*, 837 F.2d 1071, 5 USPQ2d 1596 (Fed. Cir. 1988), and *In re Jones*, 958 F.2d 347, 21 USPQ2d 1941 (Fed. Cir. 1992). In this case, it would have been obvious to one with ordinary skill in the art at the time of invention to incorporate detecting speech content in a signal as taught by Nagata et al., in the apparatus of Martino et al., in view of Slyh et al., because, an

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artisan would readily recognize that it would effectively place the microphone for higher degree of speech recognition, and would effectively suppress background noise and localize directional sound sources within a kiosk with a steerable microphone array having multiple lobes.

For the above reasons, it is believed that the rejections should be sustained.

Respectfully submitted,





Vijay B. Chawan  
Primary Examiner  
Art Unit 2654

vbc

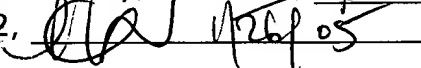
January 26, 2005

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